TELECOMMUNICATIONS ENGINEERING

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SECOND EDITION

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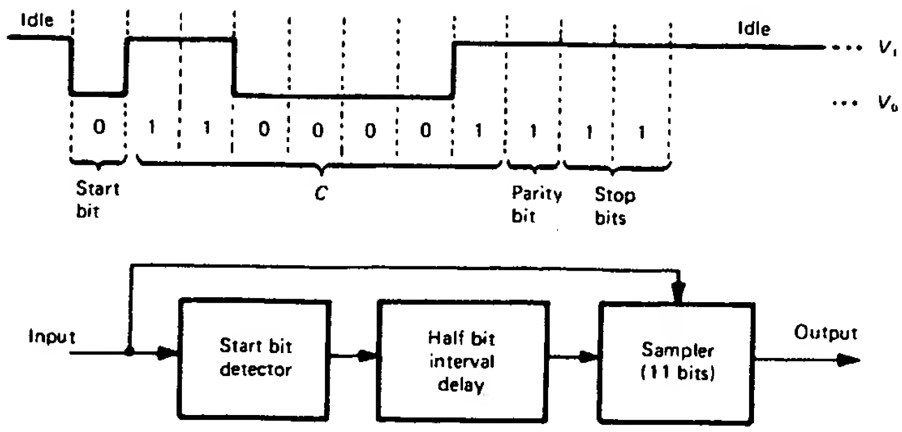


Fig. 3.28 Asynchronous transmission.

3.12 Spectral Properties of Data Signals

We must know something of the spectral properties of data signals before we can specify the most appropriate form of data transmission over telephone circuits. A typical data signal will consist of a random sequence of pulses of binary 1s and 0s. The power spectral density of such a signal was derived in Section 1.12 from its autocorrelation function. A random binary signal with pulse amplitudes of 0 or A V and pulse duration t_1 seconds has an amplitude spectrum given by

$$H(f) = At_1 \operatorname{sinc}(\pi f t_1)$$

It can be seen from Fig. 3.29 that most of the energy in the spectral envelope is confined to frequencies below $f = 1/t_1$ Hz. The bandwidth of the data signal is therefore usually approximated by the reciprocal of the pulse width.

The data spectrum, which has a component at zero frequency must be modified for transmission over a telephone circuit which usually has a bandwidth from 300 Hz to 3.4 kHz. Further, since two-way signalling is required over a single circuit, it is necessary to differentiate between transmitted and received data signals. Both these requirements are met by modulating the data signal on to an audio frequency tone. The three possible forms of modulation are AM, FM and PM.

3.13 Amplitude Shift Keying (ASK)

This is the name given to AM when used to transmit data signals. It is not normally used on telephone lines because the large variations in circuit attenuation which can occur make it difficult to fix a threshold for deciding

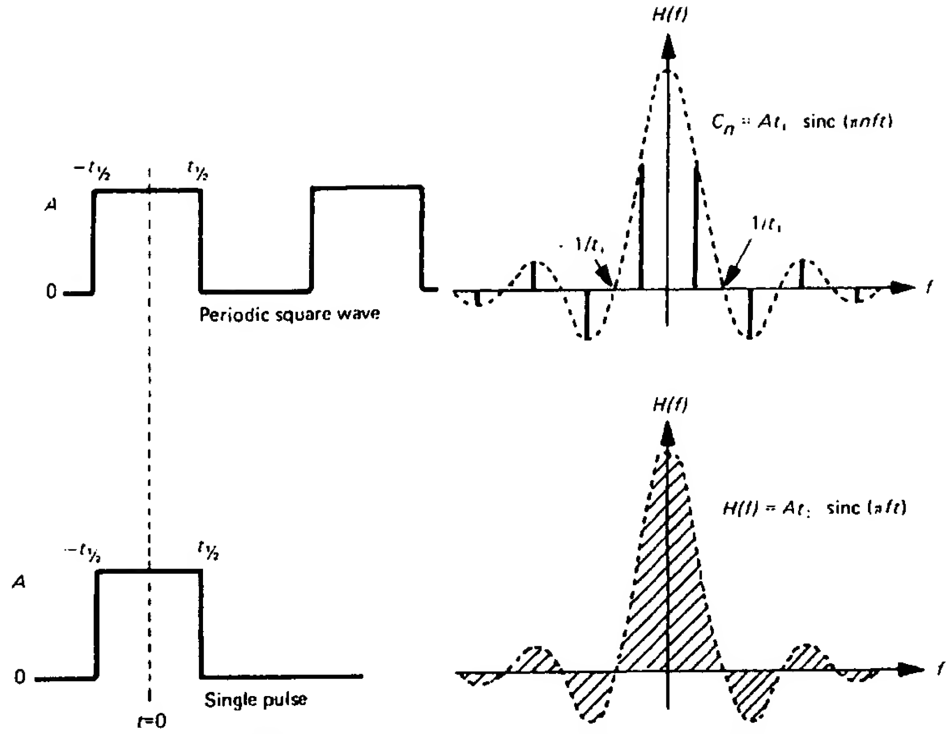


Fig. 3.29 Amplitude spectrum of data signals.

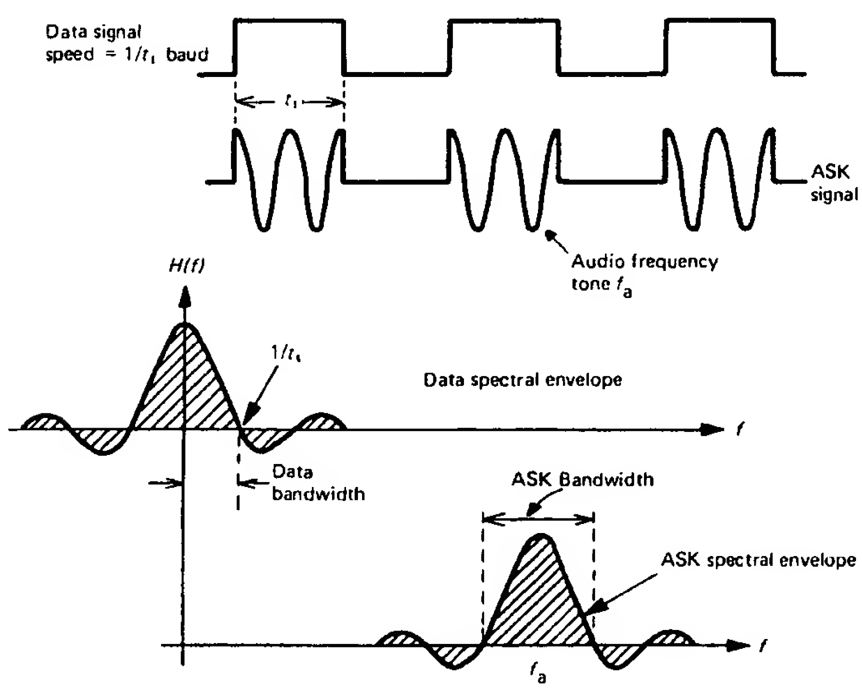


Fig. 3.30 ASK amplitude spectrum.

between binary 1 and 0. We shall, however, consider ASK in some detail because it is convenient to represent FM as the sum of two ASK signals.

The ASK signal is generated by multiplying the data signal by an audio tone. This effectively shifts the data spectrum to a centre frequency equal to that of the audio tone. The process is shown in Fig. 3.30. The bandwidth of the modulated signal is twice the bandwidth of the original data signal. This means that the original 110 baud signalling rate requires a transmission bandwidth of 220 Hz using ASK:

ASK ≡ DSBAM ≡ (carrier + upper and lower sidebands)

3.14 Frequency Shift Keying (FSK)

This is the binary equivalent of FM. In this case a binary 0 is transmitted as an audio frequency tone f_0 and a binary 1 is transmitted as a tone f_1. Hence the binary signal effectively modulates the frequency of a "carrier".

Although, strictly speaking, FSK is FM, it is more convenient to consider FSK as the sum of two ASK waveforms with different carrier frequencies. The spectrum of the FSK wave is thus the sum of the spectra of the two ASK waves.

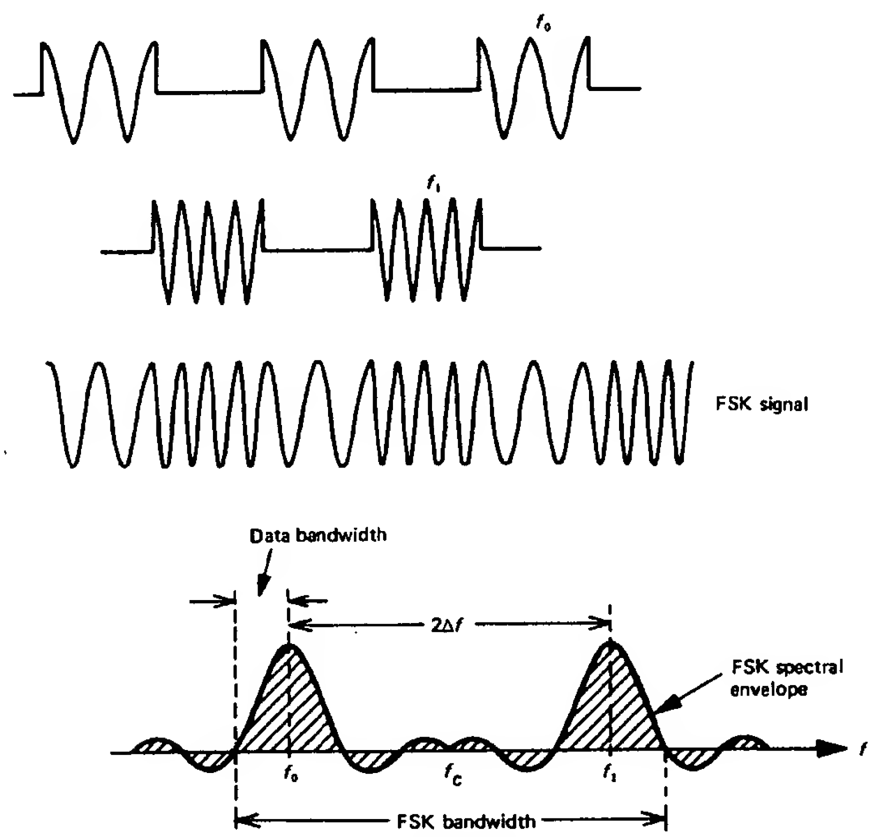
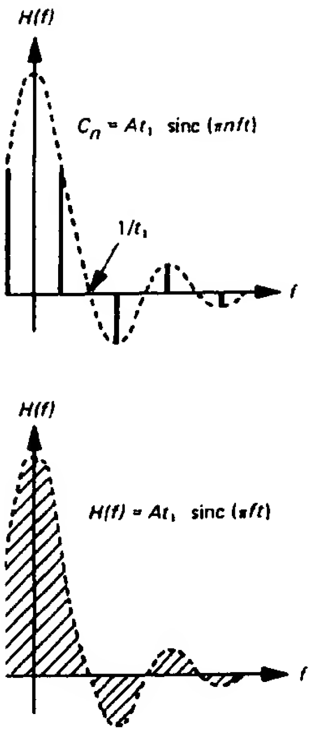


Fig. 3.31 FSK amplitude spectrum.



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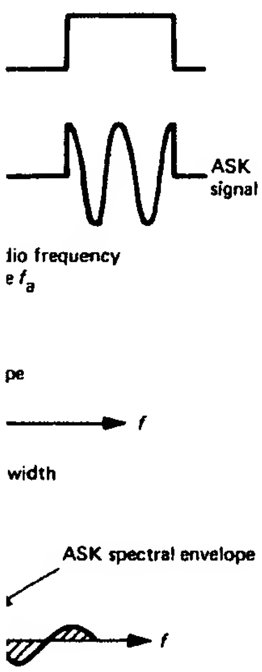


rum.

This spectrum is shown in Fig. 3.31. Using the FM analogy, it is possible to define a "carrier frequency" $f_c = f_0 + (f_1 - f_0)/2$ and a "carrier deviation" $\Delta f = (f_1 - f_0)/2$. The modulation index β is defined as $\beta = \Delta f/B$, where $B = 1/t_1$ is the bandwidth of the data signal. Using these definitions the bandwidth of the FSK signal is

$$B_{FSK} = 2B(1 + \beta) \tag{3.37}$$

This is similar to Carson's rule for continuous FM. Unlike analogue FM there is no advantage in increasing Δf beyond the value $\Delta f = B$ since the receiver only needs to differentiate between the two tones f_0 and f_1.

3.15 Phase Shift Keying (PSK)

This is the binary equivalent of PM, the binary information being transmitted either as zero phase shift or a phase shift of π radians. This is equivalent to multiplying the audio tone by either $+1$ or -1. The bandwidth is thus the same as for ASK. Since there is no dc component in the modulating signal, the carrier in the PSK spectrum will be suppressed. The equivalent modulating signal and the PSK spectral envelope are shown in Fig. 3.32. This form of PSK is sometimes referred to as binary PSK (BPSK) because the phase shift is restricted to two possible values, and it is equivalent to binary DSB-SC-AM.

3.16 Practical Data Systems

We have already indicated the reason for not using ASK for data communications on the public telephone network. The choice between FSK and PSK is

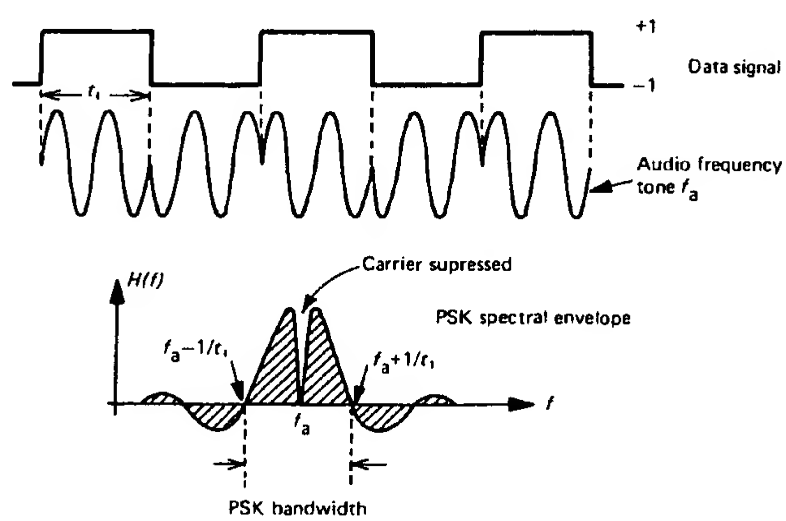


Fig. 3.32 PSK amplitude spectrum.

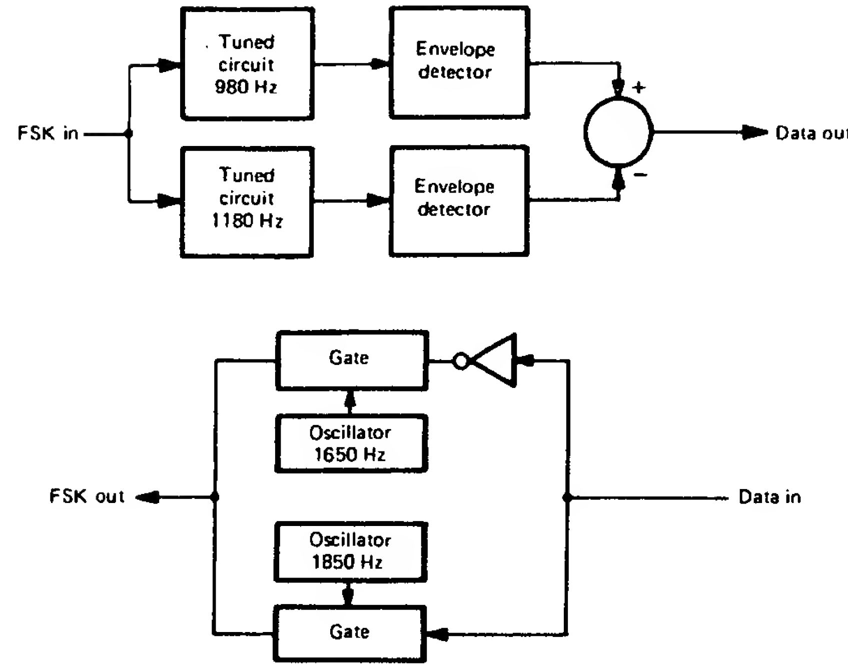


Fig. 3.33 200 baud modem.

determined by the data rate. At low data rates FSK is employed for two-way (duplex) communication. A typical system operating at a signalling rate of 200 baud uses two tones of 980 Hz and 1180 Hz for binary 1 and 0 in one direction and 1650 Hz and 1850 Hz for binary 1 and 0 in the reverse direction. The incoming FSK is separated into two tones using bandpass filters. Envelope detection is then used to reproduce the binary signal. The combined modulator/demodulator (modem) is illustrated in schematic form in Fig. 3.33.

As the data rate is increased higher carrier frequencies are required; otherwise, each data interval would contain very few cycles of carrier, which would make detection extremely difficult. There is a limit on carrier frequency imposed by the upper cut-off frequency of the telephone line. For this reason FSK is limited to signalling speeds up to 600 baud. At speeds above this PSK is employed. This type of modulation makes more efficient use of bandwidth but requires more sophisticated coherent detectors. The reference signal for coherent detection is derived from the PSK signal itself. Since the carrier is suppressed in the PSK spectrum the received waveform is first rectified to produce a component at twice carrier frequency. This component is then limited and divided by two to produce the required reference signal. The required signal processing is illustrated in Fig. 3.34.

In the public telephone network any connection between transmitter and receiver will be made via several different paths which will contain several stages of frequency multiplexing and demultiplexing. Imperfections in the various stages of modulation result in random, slowly varying, phase shifts which are

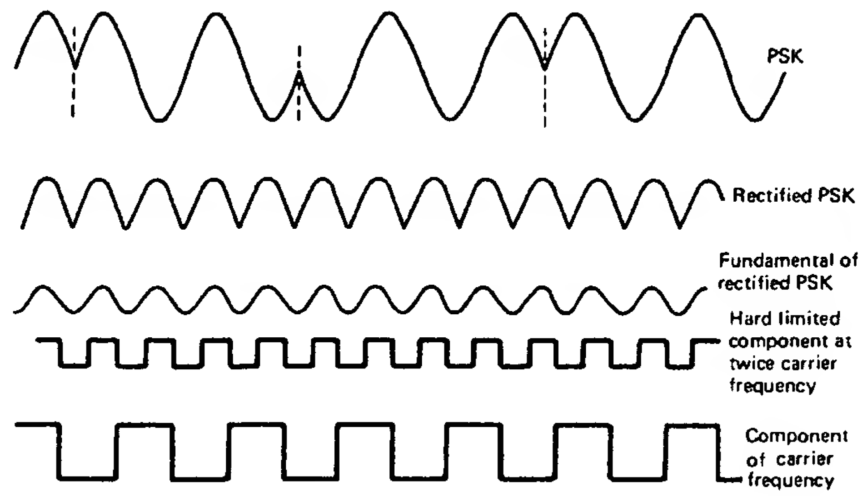


Fig. 3.34 Coherent detection of PSK.

introduced into the PSK waveform. This results in phase ambiguity at the receiver and can produce data inversion.[6] The problem is greatly reduced if differential encoding is employed.

3.17 Differential Phase Shift Keying (DPSK)

DPSK has the advantage of using the phase of the previous bit interval as the reference for the present bit interval. In order to make this possible, a binary 0 is transmitted as the same phase as the previous digit and a binary 1 is transmitted as a change of phase. The relationship between PSK and DPSK is shown in Fig. 3.35. The receiver compares the phase of the current digit with the phase of the previous digit. If they are the same the current digit is interpreted as a 0; otherwise it is interpreted as a 1. DPSK can be produced by pre-coding the data signal which then modulates the carrier as in standard PSK. If A_n is the present input to the encoder (A_n is binary) and C_{n-1} is the previous output the truth

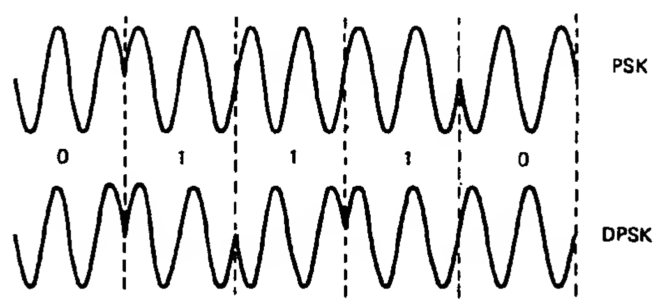
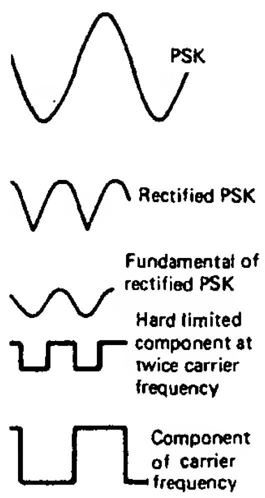


Fig. 3.35 Relationship between PSK and DPSK.



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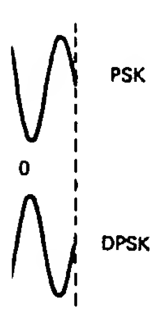

ıd DPSK.

table for the encoder is

A_n	C_{n-1}	C_n
0	0	0
0	1	1
1	0	1
1	1	0

which will be recognized as the exclusive-OR operation

$$C_n = A_n \oplus C_{n-1}$$

We have already noted that pulses can be transmitted at a rate of $1/t_1$ without mutual interference over a channel of cut-off frequency t_1 Hz, provided that the channel has a raised cosine frequency response. This applies to the unmodulated signal. A signalling rate of 1200 baud thus requires a raised cosine channel of bandwidth 1200 Hz. The PSK signal will require a bandpass channel with a raised cosine characteristic with a bandwidth of 2400 Hz. A typical PSK signal system would operate at a carrier frequency of 1.8 kHz and a signalling rate of 1200 baud. The bandwidth occupied by this waveform extends from 600 Hz to 3 kHz. Hence a data rate of 1.2 kbits/s is an upper limit for BPSK.

3.18 Advanced Modulation Methods

The data rate of BPSK is sometimes expressed as 1 bit/baud. Since the baud rate is fixed by the channel characteristics the data rate can only be increased by increasing the number of levels per pulse beyond two. If each pulse has four levels the data rate becomes 2 bits/baud and each level can produce a unique phase shift. Thus it is possible to transmit data at a rate of 2.4 kbits/s without any increase in bandwidth. It is convenient when considering multiphase PSK to represent the transmitted signal in terms of the sum of two quadrature audio frequency tones, i.e.

$$f_c(t) = a\cos 2\pi ft + b\sin 2\pi ft \tag{3.38}$$

Each of the levels in a 4-level pulse can be represented by two binary digits called dibits. Thus it is not actually necessary in practice to produce a 4-level pulse; instead the binary signal can be grouped into dibits and each dibit can be used to produce a unique phase shift in multiples of $\pi/2$. When interpreted in this way each dibit represents one pulse, i.e. the signalling rate in bauds equals half the bit rate. Table 3.3 lists the possible dibits and the values of a and b in Eqn (3.38) necessary to produce the required phase shifts. The resulting quaternary PSK can be represented on a signal space diagram of the type shown in Fig. 3.36.

The data signal is recovered from the QPSK waveform by using two coherent detectors supplied with locally generated carriers in phase quadrature. The data

Table 3.3

Dibit	Phase shift	In-phase component a	Quadrature component b
00	$\pi/4$	$+1$	$+1$
01	$3\pi/4$	-1	$+1$
11	$-3\pi/4$	-1	-1
10	$-\pi/4$	$+1$	-1

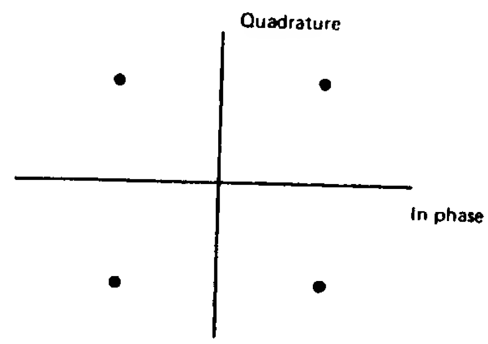

Fig. 3.36 Signal space diagram for QPSK.

rate can be increased further by increasing the number of levels of each pulse beyond four. For example, if the number of levels is increased to 16, it is possible to transmit data at a rate of 4 bits/baud. The QPSK signal was characterized by the fact that the coefficients a and b of Eqn (3.38) always have the same magnitude, thereby producing a resultant of constant amplitude and varying phase. It is also possible for a and b to have different values and the resulting signal is in fact quadrature amplitude modulation. The detection of QAM is covered in Section 11.11 in connection with the transmission of chrominance signals in the PAL colour television system.

The signal space diagram for QAM with 16-level pulses is shown in Fig. 3.37. Each individual level is represented by a unique combination of a and b in Eqn (3.38). It is possible with this system to transmit data at a rate of 4.8 kbits/s over a raised cosine bandpass channel, with a bandwidth of 2.4 kHz.

At these high data rates, intersymbol interference is a severe problem and elaborate equalization networks (transversal digital filters) are always employed. The lines used are not part of the public telephone network and are maintained to close tolerances in respect of loss and bandwidth. Such lines are often known as leased lines. The sophisticated equalization used on these lines means that signalling rates can approach the Nyquist rate, i.e. pulses can be transmitted at rates approaching 2400 baud. This means that it is possible to transmit data rates up to 9.6 kbits/s using 16-level QAM.

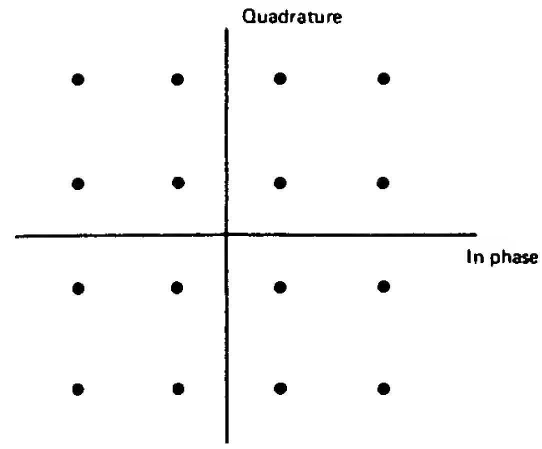

Fig. 3.37 Signal space diagram for 16-level QAM.

The leased lines referred to in the previous paragraph are basically high grade voice channels. Much higher data rates are possible on specially designed wideband links. These wideband links operate in a synchronous fashion and modern networks are adopting packet-switching techniques to maximize the efficiency of usage of these links.

3.19 Conclusions

This chapter has introduced the basic concepts of digital communications and stressed the advantages of transmission of information in digital format. This is a huge growth area in telecommunications systems engineering and is likely to remain so for the foreseeable future. All digital telephone networks, including cellular mobile telephones, are progressively being installed. These networks will gradually replace analogue systems and will provide enhanced services and increased reliability. A detailed treatment of packetized transmission is given in Chapter 13.

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